contemplate the effects of packet loss. Further, any modifications to Shlomot based upon these other references would have no reasonable expectation of success. Dealing with both compression/expansion and packet loss, as is done in the claimed invention, is a non-trivial endeavor.

CONCLUSION

In view of the foregoing, Applicants believe all claims now pending in this Application are in condition for allowance. Applicant respectfully requests reconsideration of this application as amended and issuance of a formal Notice of Allowance at an early date.

If the Examiner believes a telephone conference would expedite prosecution of this application, please telephone the undersigned at 303-571-4000.

Respectfully submitted,

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VERSION WITH MARKINGS TO SHOW CHANGES MADE

IN THE SPECIFICATION:

Please delete the last paragraph on page 4, line 32 and substitute the following paragraph therefor:

Examples of the methods described above are disclosed in "The Effect of Waveform Substitution on the Quality of PCM Packet Communications," O. J. Wasem et al., IEEE Trans. Signal Proc., vol. SP-36, no. 3, pp. 342-348, 1988.

Please delete the last three paragraphs on page 6, line 24 and substitute the following paragraph therefor:

In one embodiment, the present invention provides a method for manipulating a received sound signal to produce a sound signal. The received sound signal is received from a packet-switched network that looses some packets. In one step, a first received frame that is part of the received sound signal is received. A first signal frame corresponding to the first received frame is produced. The first signal frame is part of the sound signal. A second received frame is normally produced contiguously with the first received frame. During production of the first signal frame, it is determined that part of the second received frame is currently unavailable for production. An expanded portion is produced after determining that the part of the second received frame is currently unavailable. The first signal frame and the expanded portion are contiguous parts of the sound signal. The expanded portion corresponds to a different amount of the received sound signal than either the first or second received frames. Various aspects of the present invention are defined in the appended claims.

IN THE CLAIMS:

Please cancel claims 1-12 without prejudice to or disclaimer of the subject matter contained therein.

13. (Amended) The method of claim [1] 26, wherein use is made of an oscillator model for extracting signal segments from the first signal frame [**used when**

manipulating the lengths of said received signal frames], the oscillator model including a codebook in which vectors of samples forms different states, or entries, in the codebook, the codebook storing a corresponding signal segment for each state.

14. (Amended) The method of claim 13, wherein the second-listed producing step comprises a step of [**said time expansion of a signal frame is performed by**] matching a true state of a trailing part of [**such**] the first signal frame with said states in said codebook, and reading out a signal segment from said codebook that corresponds to the state having been matched with said true state.

15. (Amended) The method of claim 13, wherein said signal segments of said codebook have variable lengths, each signal segment forming a trailing part of a signal frame, thereby enabling continuous transition from the [**time**] expanded portion [**signal frame**] to a consecutive signal frame.

16. (As Filed) The method of claim 13, wherein time delays between said states in said codebook are incremental delays with a resolution of a fraction of a time between two samples.

17. (As Filed) The method of claim 14, wherein the states and the corresponding segments of said codebook are scaled in order to improve the matching with said true state.

18. (As Filed) The method of claim 14, wherein merging of said true state is performed with the matching state of said codebook.

19. (Amended) The method of claim 14, wherein the second-listed producing step [**said time expansion additionally**] involves performing the corresponding operations with respect to a heading part of [**a**] the second signal frame being consecutive to the [**time**] expanded portion [**signal frame**].

20. (Amended) The method of claim [1] 26, wherein said first signal frame[**, which length is to be manipulated,**] is either a sound signal frame resulting from a complete decoding operation of the first received frame [**a data packet**],

or an intermediate time-domain signal frame resulting from a partial decoding operation of the first received frame [**a data packet**].

21. (Amended) The method of claim [1] 26, including the step of using an oscillator model, which oscillator model includes a codebook in which vectors of samples of a received digitized sound signal forms different states, or entries, in the codebook, the codebook storing a corresponding signal segment for each state.

Please delete claims 22-25 without disclaimer of or prejudice to the subject matter contained therein.

26. (New) A method for manipulating a received sound signal to produce a sound signal, wherein the received sound signal is received from a packet-switched network that looses some packets, the method comprising steps of:

receiving a first received frame that is part of the received sound signal;

producing a first signal frame corresponding to the first received frame, wherein:

the first signal frame is part of the sound signal, and

a second received frame is normally produced contiguously with the first received frame;

determining after beginning the first-listed producing step that at least part of the second received frame is currently unavailable for production; and

producing an expanded portion after the determining step, wherein:

the first signal frame and the expanded portion are contiguous parts of the sound signal, and

the expanded portion that corresponds to a different amount of the received sound signal than either the first or second received frames.

27. (New) The method of claim 26, wherein the expanded portion is selected from the first signal frame based, at least in part, upon measures of periodicity.

28. (New) The method of claim 26, wherein the determining step comprises a step of determining near the end of production of the first signal frame if the part of the second received frame is currently unavailable for production.

29. (New) The method of claim 26, further comprising steps of:

determining after beginning the second-listed producing step that the second received frame is still unavailable for production;

producing a second expanded portion after the immediately-preceding determining step, wherein the expanded portion and the second expanded portion are contiguous parts of the sound signal.

30. (New) The method of claim 26, wherein:

a playback time of the expanded portion is variable, and

the playback time is selected based, at least in part, upon the sound signal.

31. (New) The method of claim 26, wherein:

the first signal frame includes a plurality of sound samples, and

the expanded portion is determined with a time resolution finer than a sample rate of the plurality of sound samples.

32. (New) The method of claim 26, further comprising a step of producing a second expanded portion based, at least in part, on some of the second received frame, wherein the expanded portion and second expanded portion are contiguous parts of the sound signal.

33. (New) The method of claim 26, further comprising a step of merging the expanded portion and a contiguous, subsequent, portion of the sound signal using a periodicity measure, whereby any audible discontinuities between the expanded portion and second expanded portion are reduced.

34. (New) The method of claim 26, wherein the signal frame corresponds to a plurality of received frames.

35. (New) The method of claim 26, further comprising a step of merging the expanded portion and a contiguous, subsequent, portion of the sound signal based, at least in part, on overlap-add, wherein a time shift of the first signal frame and expanded portion is optimized based, at least in part, on correlation.

36. (New) The method of claim 26, further comprising steps of:

measuring overload of a jitter buffer;

discarding some of the second received frame based, at least in part, on the overload; and

merging a preceding and a subsequent portions of the sound signal after the discarding step.

37. (New) The method of claim 36, further comprising steps of:

determining if a signal fitting criteria between the preceding and subsequent portions is fulfilled; and

performing the discarding step only with the immediately-preceding determining step is fulfilled.

38. (New) The method of claim 36, wherein a length of the some of the second received frame is based, at least in part, on the sound signal.

39. (New) The method of claim 36, wherein the some of the second received frame comprises a plurality of sub-portions that are sequentially discarded.

40. (New) The method of claim 36, wherein:

the merging step is based, at least in part, on overlap-add, and

any time-shift of the preceding and subsequent portions is optimized based, at least in part, on a measure of periodicity.

41. (New) A computer-readable medium having computer-executable instructions for performing the computer-implementable method of claim 26.

42. (New) A computer system adapted to perform the computer-implementable method of claim 26.

43. (New) A method for manipulating a received sound signal to produce a sound signal, wherein the received sound signal is received from a packet-switched network that looses some packets, the method comprising steps of:

receiving a first received frame that is part of the received sound signal;

producing a first signal frame corresponding to the first received frame, wherein the first signal frame is part of the sound signal;

determining after beginning the first-listed producing step that part of the second received frame currently unavailable for production; and

producing an expanded portion after the determining step, wherein:

the expanded portion and a second signal frame are contiguous parts of the sound signal,

the first signal frame and the second signal frame would be contiguous parts of the sound signal in situations where the part of the second received frame is available for production, and

the expanded portion is a different size than either the first or second received frames.

44. (New) A method for manipulating a received sound signal to produce a sound signal, wherein the received sound signal is received from a packet-switched network that looses some packets, the method comprising steps of:

receiving a first received frame that is part of the received sound signal;

producing a first signal frame corresponding to the first received frame, wherein:

the first signal frame is part of the sound signal, and

a second received frame is produced contiguously with the first received frame when the second received frame is available;

determining after beginning the first-listed producing step that part of the second received frame is currently unavailable for production; and

producing an expanded portion after the determining step, wherein:

the first signal frame and the expanded portion are contiguous parts of the sound signal,

the expanded portion replaces at least some of the second received frame, and

the expanded portion is a different size than either the first or second received frames.